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(54) METHODE AMELIOREE D'UTILISATION D'UN ANNULATEUR D'ECHO POUR TELEPHONE A HAUT-PARLEUR EN DUPLEX TOTAL

(54) IMPROVED METHOD OF OPERATING A FULL DUPLEX SPEAKERPHONE ECHO CANCELLER
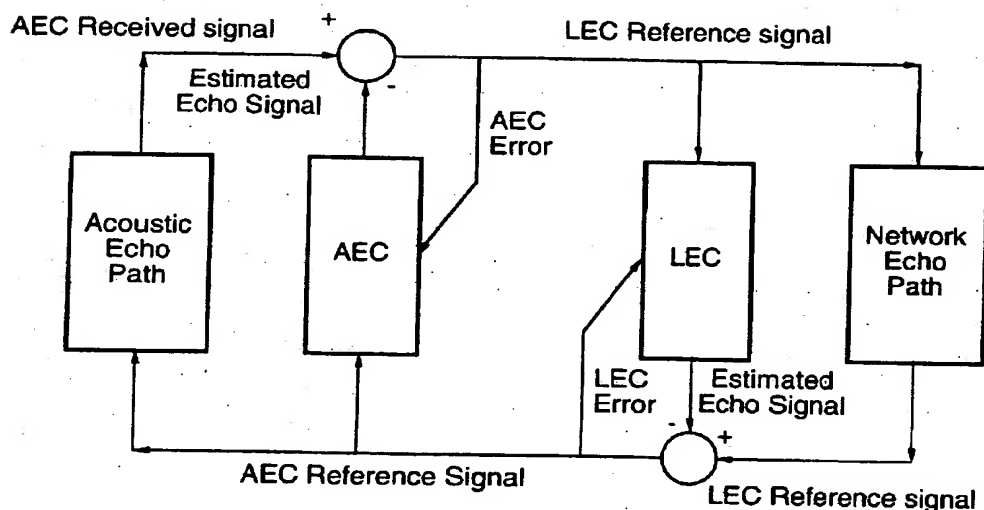
(57)
A method of improving convergence of an echo canceller in a full duplex speakerphone, wherein the echo canceller includes LEC (Line Echo Canceller) and AEC (Acoustic Echo Canceller) portions, comprising the steps of capturing LEC coefficients during operation, storing the coefficients, and utilizing the stored coefficients as default values during start-up of a subsequent call. The method of the present invention reduces the overall convergence time of the echo canceller by alleviating the requirement to wait for a suitable reference signal in order to converge the LEC.

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ABSTRACT

A method of improving convergence of an echo canceller in a full duplex speakerphone, wherein the echo canceller includes LEC (Line Echo Canceller) and

5   AEC (Acoustic Echo Canceller) portions, comprising the steps of capturing LEC coefficients during operation, storing the coefficients, and utilizing the stored coefficients as default values during start-up of a subsequent call. The method of the present invention reduces the overall convergence time of the echo canceller by alleviating the requirement to wait for a suitable reference signal in order to converge

10   the LEC.

IMPROVED METHOD OF OPERATING A FULL DUPLEX SPEAKERPHONE ECHO CANCELLER

Field of the Invention

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The present invention relates in general to speakerphones and more particularly to an improved method of operating an echo canceller in a full duplex speakerphone connected to an analog line, where the near end hybrid characteristic is the same for each connection.

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Background of the Invention

One of the most important performance indicators for full duplex speakerphones is convergence time (i.e. the time required by the echo cancellers within the speakerphone to reach an acceptable level of cancellation). The convergence time of the speakerphone depends both on internal Line Echo Canceller (LEC) and Acoustic Echo Canceller (AEC) convergence times. In order to converge quickly and properly, a speakerphone echo canceller requires a reference signal with correct stochastic properties. At the beginning of a call (Start-up), the reference signal is usually not sufficiently stochastic (e.g. the line signal typically comprises narrow band tones such as dial tone) or speech is not present, so that echo cancellation is unable to commence immediately. In such situations the speakerphone loop may remain unstable for a noticeable period of time. This can result in feedback or "howling" of the speakerphone during start-up, especially when the speaker volume is high.

In order to prevent such feedback, it is an objective of speakerphone design to ensure that the echo cancellers (LEC and AEC) converge rapidly to the correct echo path models at start-up. Otherwise, the speaker volumes must be reduced during start-up, which may be annoying to a user.

Since the LEC usually models an echo path that is reasonable constant, and the
AEC often has to track frequent changes in the echo path, it is advantageous if the
LEC filter adapts quickly to the correct model and remains stable while the AEC
undergoes convergence.

According to one prior art approach to reducing the problem of feedback
during speakerphone start-up, howling detection has been used (see ITU-T
Recommendation G.168) in combination with gain control. According to this
approach, the speaker volume (or loop gain) is reduced when howling is detected. A
drawback of this approach is that the gain switching is often audible which may be
annoying to the user.

Another prior art solution involves operating the speakerphone in a half duplex
mode on start-up in order to prevent howling and echo from interfering with
communication. The speakerphone remains in the half-duplex mode until the LEC
adapts sufficiently to ensure echo cancellation. A drawback of this approach is that the
speakerphone sometimes stays in the half-duplex mode for a long time, making
communication between telephone parties difficult or impossible.

Yet another prior art solution involves forcing the speakerphone to start
operation at a predetermined "acceptable" low volume level which guarantees
stability in the audio loop, and then gradually increasing the volume as convergence
of the echo canceller is achieved. A drawback of this approach is that the volume
adjustment is often noticeable to the user.

Summary of the Invention

According to the present invention, a method is provided for improving the
start-up convergence time of the LEC filter, thereby resulting in a total reduced
convergence time for the speakerphone. This method is based on capturing the LEC
coefficients once the LEC has converged, and saving them as the default coefficients

for the next call. As a result, the echo canceling algorithm does not have to wait for a suitable reference signal to commence convergence. At start-up, the echo canceller immediately begins canceling the line echo, based on the previously stored LEC coefficients, thereby assisting the AEC algorithm by eliminating residual line echo

5   from the acoustic signal which the AEC algorithm is required to converge to, and initially making the speakerphone loop more stable.

Brief Description of the Drawings

10   A detailed description of the prior art and of a preferred embodiment of the invention is provided herein below with reference to the following drawings, in which:

Figure 1 is a block diagram of a prior art speakerphone echo canceller

15   structure: and

Figure 2 is a flow chart showing the steps of the echo cancellation method according to the present invention.

20   Detailed Description of Prior Art and Preferred Embodiment

As discussed briefly above, a speakerphone echo canceller comprises two adaptive filters which attempt to converge to two different echo models (acoustic and network echo) at the same time. As a result, speakerphones can easily become

25   unstable, especially during start-up.

A traditional speakerphone echo canceller is shown in Figure 1, wherein essential speakerphone components which are not related to echo cancellation have been omitted for clarity (e.g. double talk detector, non-linear processor, etc.) and are

30   not addressed herein since they are not germane to the invention. The echo canceller attempts to model the transfer function of the echo path by means of an LEC filter and an AEC filter. The received signal (line or acoustic) is applied to the input of each

filter (LEC and AEC) and to the associated echo path (network or acoustic) such that the estimated echo can be canceled by simply subtracting the signal which passes through each echo canceller from the received signal. If the transfer function of the model of the echo path is exactly the same as the transfer function of the echo path,

5   the echo signal component is completely canceled (i.e. the error signal will be zero). The error signal is used for adaptation, so that the echo canceller converges to the correct transfer function, as discussed briefly above.

Typically, an algorithm such as the NLMS (Normalized-Least-Mean-Squared)

10   algorithm is used to approximate the echo path (see "C261(UNIC) DSP Re-engineering and Performance Report" Mitel Semiconductor, Document No. C261AP13, Oct. 21, 1996).

From Figure 1 it will be appreciated that the residual echo after imperfect

15   cancellation by the LEC will pass to the AEC reference signal. Since this residual echo is not correlated to the AEC received signal, this can cause the AEC filter to diverge. The extent to which AEC filter diverges depends on the level of the residual line echo. If the line echo is sufficiently canceled, its effect on the AEC behavior will be negligible.

20

Echo Return Loss Enhancement (ERLE) is an indicator of the amount of echo removed by an echo canceller. The ERLE is defined as:

$$ERLE(dB) = 10\log_{10}[Power(ReceivedSignal)/Power(ErrorSignal)];$$

25

A generally acceptable LEC convergence time requires that the echo canceller achieve 27dB of ERLE in 0.5 sec (in ideal conditions).

Since the telephone is always connected to the same local loop (i.e. to the

30   near-end Central Office (CO) or PBX), the impedance of the local loop remains the same for each call and consequently the near-end echoes remain fairly constant, from call to call. Accordingly, according to the present invention the local loop echo

coefficients can be stored and re-used from call to call, thereby improving the start-up ERLE of the LEC.

5   Thus, with reference to the flowchart of Figure 2, after start-up of the echo canceller (Step 200), any previously stored default LEC coefficients are loaded into the LEC and the LEC begins convergence using the well known NLMS algorithm (or other). On initial power-up of the speakerphone (i.e. prior to placing the first call), the initial coefficients are zero. Thus, the first call after power-up will always be a "training" call that results in capturing a suitable set of default coefficients for future
10   calls. Next, the algorithm according to the present invention is executed (referred to herein as Call – step 201). The signal levels of the LEC received signal and error signal are detected (step 203) and the ERLE is calculated using the formula set forth above (step 205). When a predetermined ERLE threshold level (Th) is reached (e.g. at least 24dB of echo is canceled), as calculated at step 207, and provided that the best
15   LEC coefficients have not been previously saved during the call-in-progress (step 209), then the LEC coefficients of the near echo are saved (step 211). Convergence of the AEC then proceeds as per usual and the call is completed (step 213). Once saved, the default coefficients are not be recalculated again for the duration of the call (i.e. a YES decision at step 209). However, the LEC default coefficients will be calculated
20   once per each call to ensure the best default set is captured for the next call.

          At start-up of the next call, the previously stored LEC coefficients are retrieved and used as the default coefficient set for the LEC (step 200), instead of starting from zero.

25

          The following pseudo code illustrates the principles of the inventive method in greater detail:

Power-up: Default_coefficients = [000...0];
30
Start_Call: LEC_coefficients = Default_coefficients;
Call:
          Execute LEC algorithm;
          Calculate power level of received signal ;

```
            Calculate power level of error signal;
            If (ERLE > Threshold) AND ( Best default set not saved)
                Save near echo coefficients
            If Not(End of the Call) Go to Call:
5           If New Call Go to Start_Call;
```

Thus, according to the algorithm or method of the present invention. each call subsequent to the initial power-up "training" call with default coefficients that model the network echo path and guarantee small LEC error. This improves the training and

10 tracking characteristic of the AEC and eliminates the feedback during start-up. The best results will be achieved when the training call uses a handset since there is no AEC-LEC loop instability and the LEC can therefore converge quickly.

Other embodiments and applications of the invention are possible. For

15 example, this algorithm with some variations may also be implemented for the AEC filter to capture the acoustic feedback through the plastic, which will be constant for the specific phone design. Although a threshold ERLE value of 24 dB is disclosed herein, the threshold value may be varied to provide optimum performance for any particular application. All such variations and modifications are believed to be within

20 the sphere and scope of the invention as set forth in the claims appended hereto.

We claim:

1.   A method of operating a full duplex speakerphone having a line echo canceller
which receives a reference signal and converges to an estimated echo signal of a
received line signal via feedback of an error signal. said echo canceller being
5   characterized by predetermined filter coefficients when converged. said method
comprising the steps of:

        a) detecting power levels of said received line signal and said error signal
during a call in progress;

10

        b) calculating an Echo Return Loss Enhancement value based on said power
levels of said received line signal and said error signal; and

        c) in the event said Echo Return Loss Enhancement value exceeds a
15   predetermined amount then storing said predetermined filter coefficients for use by
said echo canceller during calls subsequent to said call in progress. and otherwise re-
executing steps a) to c).

2.   The method of claim 1. wherein said Echo Return Loss Enhancement value is
20   expressed in decibels as
$ERLE(dB)=10log_{10}[Power(ReceivedSignal)/Power(ErrorSignal)]$.

3.   The method of claim 2. wherein said predetermined amount is at least 24dB

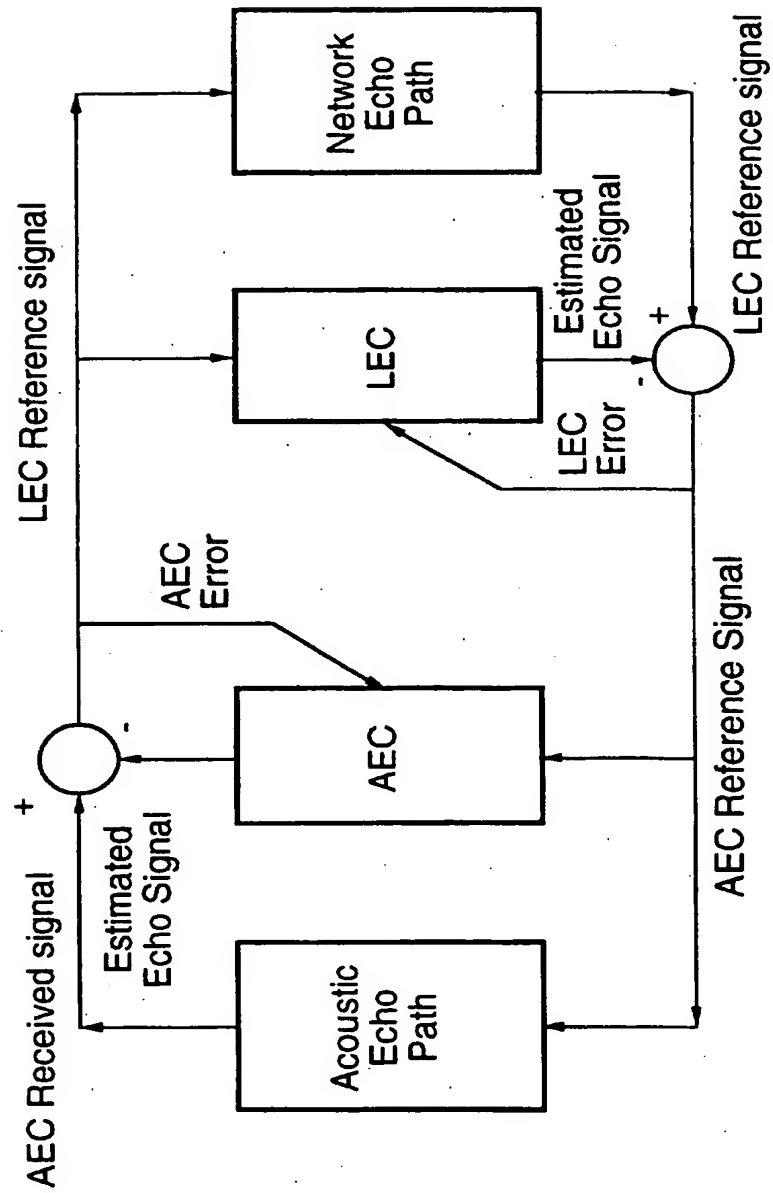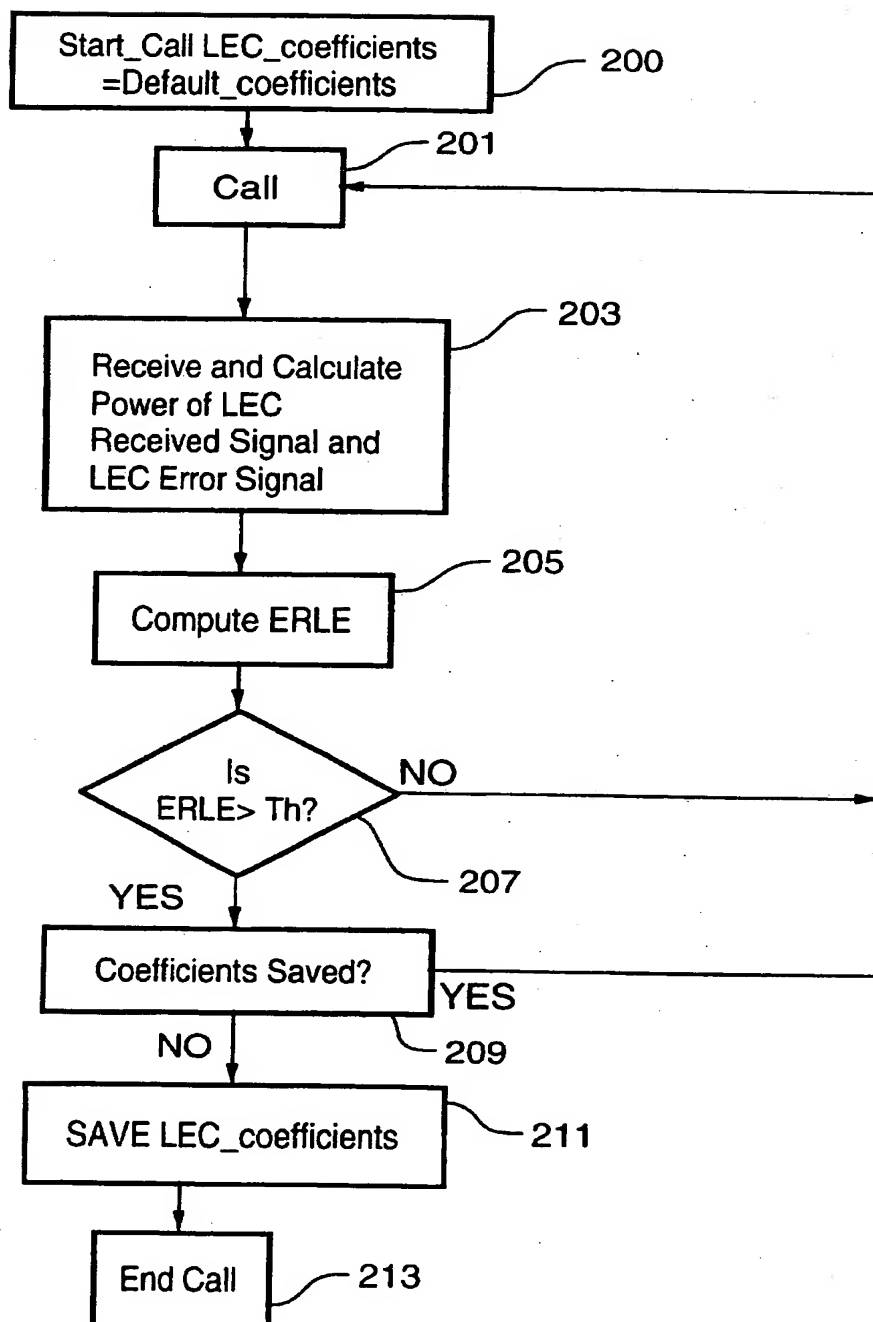25   4.   The method of claim 2. wherein said predetermined amount is at least 27dB.

FIG.1

Start_Call LEC_coefficients =Default_coefficients — 200

Call — 201

Receive and Calculate Power of LEC Received Signal and LEC Error Signal — 203

Compute ERLE — 205

Is ERLE> Th? — 207
NO
YES

Coefficients Saved? — 209
YES
NO

SAVE LEC_coefficients — 211

End Call — 213

FIG.2